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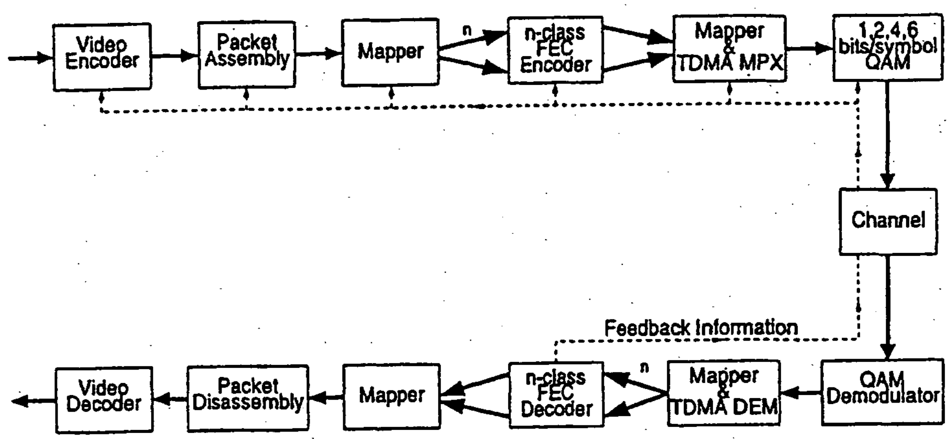
(54) Title: BURST–BY–BURST ADAPTIVE SINGLE–CARRIER DATA TRANSMISSION



(57) Abstract

The performance benefits of burst–by–burst adaptive modulation are studied, employing a higher–order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source–signal representation quality – such as video or audio quality – versus channel quality relationship can be established, resulting in a near–unimpaired multimedia source–signal quality right across the operating channel Signal–to–Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source–signal representation quality – such as video or audio quality – by automatically adjusting the achievable bitrate and the associated multimedia source–signal representation quality in order to match the channel quality experienced. This is achieved on a near–instantaneous or burst–by–burst adaptive basis under given propagation conditions in order to cater for the effects of path–loss, fast–fading, slow–fading, dispersion, co–channel interference, etc. Furthermore, when the mobile is roaming in a hostile out–doors – or even hilly terrain – propagation environment, typically low–order, low–rate modem modes are invoked, while in benign indoor environments predominantly the high–rate, high source–signal representation quality modes are employed.

Title of the Invention

Burst-by-burst Adaptive Single-carrier Data Transmission

1 Background of the Invention

The invention relates to data transmission, more specifically to transmission in packets or bursts.

In contrast to the *burst-by-burst reconfigurable wideband* multimedia transceivers described in this document, the term *statically reconfigurable* found in this context in the literature refers to multimedia transceivers that cannot be near-instantaneously reconfigured. More explicitly, the previously proposed *statically reconfigurable* video transceivers were reconfigured on a long-term basis under the base station's control, invoking for example in the central cell region - where benign channel conditions prevail - a less robust, but high-throughput modulation mode, such as 4 bit/symbol Quadrature Amplitude Modulation (16QAM), which was capable of transmitting a quadruple number of bits and hence ensured a better video quality. By contrast, a robust, but low-throughput modulation mode, such as 1 bit/symbol Binary Phase Shift Keying (BPSK) can be employed near the edge of the propagation cell, where hostile propagation conditions prevail. This prevented a premature hand-over at the cost of a reduced video quality.

The philosophy of the fixed, but programable-rate proprietary video codecs and statically reconfigurable multi-mode video transceivers presented by Streit et al in References [1]-[4] was that irrespective of the video motion activity experienced, the specially designed video codecs generated a constant number of bits per video frame. For example, for videophony over the second-generation Global System of Mobile Communications known as the GSM system at 13 kbps and assuming a video scanning rate of 10 frames/s, 1300 bits per video frame have to be generated. Specifically, two families of video codecs were designed, one refraining from using error-sensitive run-length coding techniques and exhibiting the highest possible error resilience and another, aiming for the highest possible compression ratio. This fixed-rate approach had the advantage of requiring no adaptive feedback controlled bitrate fluctuation smoothing buffering and hence exhibited no objectionable video latency or delay. Furthermore, these video codecs were amenable to video telephony over fixed-rate second-generation mobile radio systems, such as the GSM.

The fixed bitrate of the above proprietary video codecs is in contrast to existing standard video codecs, such as the Motion Pictures Expert Group codecs known as MPEG1 and MPEG2 or the ITU's H.263 codec, where the time-variant video motion activity and the variable-length coding techniques employed

techniques by Matsuoka *et al* [12] as well as Goldsmith *et al.*[13]-[15].

In the narrow-band channel environment, the quality of the channel was determined by the short term Signal to Noise Ratio (SNR) of the received burst, which was then used as a criterion in order to choose the appropriate modulation mode for the transmitter, based on a list of switching threshold levels, l_n [9, 10]. However, in a wideband environment, this criterion is not an accurate measure for judging the quality of the channel, where the existence of multi-path components produces not only power attenuation of the transmission burst, but also intersymbol interference. Subsequently, a new criterion has to be defined to estimate the wideband channel quality in order to choose the appropriate modulation scheme.

2 Summary of the Invention

Particular and preferred aspects of the invention are set out in the accompanying independent and dependent claims. Features of the dependent claims may be combined with those of the independent claims as appropriate and in combinations other than those explicitly set out in the claims.

The performance benefits of burst-by-burst adaptive modulation are described, employing a higher-order modulation scheme, when the channel is favourable, in order to increase the system's bits per symbol capacity and conversely, invoking a more robust, lower order modulation scheme, when the channel exhibits inferior channel quality. It is shown that due to the described adaptive modem mode switching regime a seamless multimedia source-signal representation quality - such as video or audio quality - versus channel quality relationship can be established, resulting in a near-unimpaired multimedia source-signal quality right across the operating channel Signal-to-Noise Ratio (SNR) range. The main advantage of the described technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This can achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when a mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

Channel Impulse Response (CIR) estimate derived using the channel sounding midamble and consequently, the signal to noise and residual interference ratio at the output of the equalizer is calculated. We note, however that the above adaptive methodology can also be extended to employing burst-by-burst adaptive channel coding associated with different-strength error correction codecs in the different transceiver modes of operation.

4.3 Channel quality metrics

The most reliable channel quality estimate is the bit error rate (BER), since it reflects the channel quality, irrespective of the source or the nature of the quality degradation.

Firstly, the BER can be estimated with a certain granularity or accuracy, provided that the system entails a channel decoder or - synonymously - Forward Error Correction (FEC) decoder employing algebraic decoding.

Secondly, if the system contains a soft-in-soft-out (SISO) channel decoder, the BER can be estimated with the aid of the Logarithmic Likelihood Ratio (LLR), evaluated either at the input or the output of the channel decoder. A particularly attractive way of invoking LLRs is employing powerful turbo codecs, which provide a reliable indication of the confidence associated with a particular bit decision in the context of LLRs. The LLR is defined as the ratio of the probabilities of a specific bit being binary zero or one. Again, this measure can be evaluated at both the input and the output of the turbo channel codecs and both of them can be used for channel quality estimation.

Thirdly, in the event that no channel encoder / decoder (codec) is used in the system, the channel quality expressed in terms of the BER can be estimated with the aid of the mean-squared error (MSE) at the output of the channel equaliser or the closely related metric, the Pseudo-Signal-to-noise-ratio (Pseudo-SNR). The MSE or pseudo-SNR at the output of the channel equaliser have the important advantage that they are capable of quantifying the severity of the inter-symbol-interference (ISI) and/or Co-channel Interference experienced, in other words quantifying the Signal to Interference plus Noise Ratio (SINR).

4.3.1 Pseudo-SNR Embodiment

A specific embodiment based on the above-mentioned pseudo-SNR is now described in more depth. Employing the pseudo-SNR has the advantage that it is generally applicable, regardless of whether or not a channel codec is present.

We found that the residual channel-induced inter-symbol-interference (ISI) at the output of the decision feedback equaliser (DFE) is near-Gaussian distributed and that if there are no decision feedback errors,

Parameter	Value
Carrier Frequency	1.9GHz
Vehicular Speed	30mph
Doppler frequency	85Hz
Normalised Doppler frequency	3.27×10^{-5}
Channel type	COST 207 Typical Urban (see Figure 3)
Number of paths in channel	4
Data modulation	Adaptive QAM (BPSK, 4-QAM, 16-QAM, 64-QAM)
Receiver type	Decision Feedback Equalizer Number of Forward Filter Taps = 35 Number of Backward Filter Taps = 7

Table 1: Modulation and channel parameters

performance degradation is fairly minor for packet dropping or frame error rates (FER) below about 5%. These packet dropping events are signalled to the remote decoder by superimposing a strongly protected one-bit packet acknowledgement flag on the reverse-direction packet, as outlined in [5]. In the embodiment we also invoked the adaptive rate control and packetisation algorithm of [5], supporting constant Baud-rate operation.

As a specific example of the burst-by-burst adaptive nultimedia system we used 176x144 pixel so-called QCIF-resolution, 30 frames/s video sequences encoded at bitrates resulting in high perceptual video quality, in order to demonstrate the performance advantages of the video transceiver. Table 1 shows the modulation- and channel parameters employed, noting again that the associated principles are applicable in the context of a whole range of other system parameters. The COST207 four-path typical urban (TU) channel model was used in quantifying the associated system performance and its impulse response is portrayed in Figure 3. As an example, we used the Pan-European FRAMES proposal as the basis for our wideband transmission system, the frame structure of which is shown in Figure 4. Employing the FRAMES Mode A1 (FMA1) so-called non-spread data burst mode required a system bandwidth of 3.9MHz, when assuming a modulation excess bandwidth of 50%. A range of other system parameters are shown in Table 2.

The specific example of the video transceiver - which is used to demonstrate the advantages of the system concept - is based on the H.263 video codec. The video coded bitstream was protected by binary Bose-

Features	Multi-rate System			
Mode	BPSK	4QAM	16QAM	64QAM
Bits/Symbol	1	2	4	6
FEC	Near Half-rate BCH			
Transmission bitrate (kbit/s)	148.2	296.4	592.8	889.3
Unprotected bitrate (kbit/s)	75.8	151.7	303.4	456.1
Effective Video-rate (kbit/s)	67.0	141.7	292.1	446.4
Video fr. rate (Hz)	30			

Table 3: Operational-mode specific transceiver parameters

by-burst adaptive wide-band modem. Figure 5 demonstrates how the burst-by-burst adaptive modem changes its modulation modes every transmission burst, ie every 4.615 ms, based on the fluctuating pseudo-SNR. The right-hand-side vertical axis indicates the associated number of bits per symbol.

By changing to more robust modulation schemes automatically, when the channel quality is reduced allows the packet loss ratio, or synonymously, the FER, to be reduced, which results in increased perceived video quality. In order to judge the benefits of burst-by-burst adaptive modulation we considered two scenarios. In the first scheme the adaptive modem always chose the perfectly estimated AQAM modulation mode, in order to provide a maximum upper bound performance. In the second scenario the modulation mode was based upon the perfectly estimated AQAM modulation mode for the previous burst, which corresponded to a delay of one Time Division Multiple Access (TDMA) frame duration of 4.615ms. This second scenario represents a practical burst-by-burst adaptive modem, where the one-frame channel quality estimation latency is due to superimposing the receiver's perceived channel quality on a reverse-direction packet, for informing the transmitter concerning the best mode to be used.

The probability of the adaptive modem using each modulation mode for a particular average channel SNR is portrayed in Figure 6 in terms of the associated modem mode probability density functions (PDFs). It can be seen at high channel SNRs that the modem mainly uses the 64QAM modulation mode, while at low channel SNRs the BPSK mode is the most prevalent one.

Figure 7 shows the transmission FER (or packet loss ratio) versus channel SNR for the 1, 2, 4 and 6 bit/symbol fixed modulation schemes, as well as for the ideal and for the one-frame delayed realistic

SNRs, as demonstrated in Figure 6.

Having shown the effect of the burst-by-burst adaptive modem on the transmission FER and effective bitrate, let us now demonstrate these effects on the decoded video quality, measured in terms of the Peak Signal-to-Noise Ratio (PSNR). Figure 10 shows the decoded video quality in terms of PSNR versus channel SNR for both the ideal and realistic adaptive modem, and for the four modes of the statically configured multi-mode modem. It can be seen that - as expected - the ideal adaptive modem, which always selects the perfect modulation modes, has a better or similar video quality for the whole range of channel SNRs. For the statically configured multi-mode scheme the video quality degrades, when the system switches from a higher-order to a lower-order modulation mode. The ideal adaptive modem however smoothes out the sudden loss of video quality, as the channel degrades. The non-ideal adaptive modem has a slightly lower video quality performance, than the ideal adaptive modem, especially at medium SNRs, since it sometimes selects the incorrect modulation mode due to the estimation delay. This can inflict video packet loss and/or a reduction of the effective video bitrate, which in turn reduces the video quality.

The difference between the ideal burst-by-adaptive modem, using ideal channel estimation and the non-ideal, realistic burst-by-burst adaptive modem, employing a non-ideal delayed channel estimation can be seen more clearly in Figure 11 for a range of video sequences. Observe that at high and low channel SNRs the video quality performance is similar for the ideal and non-ideal adaptive modems. This is because the channel estimation delay has little effect, since at low or high channel SNRs the adaptive modem is in either BPSK or 64QAM mode for the majority of the time. More explicitly, the channel quality of a transmission frame is almost always the same as that of the next; and hence the delay has little effect at low and high SNRs.

The video quality versus channel quality trade-offs can be more explicitly observed in Figure 12. This figure portrays the decoded video quality in PSNR versus the packet loss ratio or transmission FER. The ideal and practical adaptive modem performance is plotted against that of the four fixed modulation schemes in the figure. It can been seen that the adaptive modems' video quality degrades from that achieved by the error-free 64QAM modem towards the BPSK modem performance as the packet loss ratio increases. The practical adaptive modems' near constant FER performance of 3% at medium SNRs can be clearly seen in the figure, which is associated with the reduced PSNRs of the various modem modes, while having only minor channel error-induced impairments.

5 Summary

The above-described burst-by-burst adaptive multimedia transceiver concept exhibits substantial advantages in comparison to conventional fixed-mode or statically reconfigurable transceivers, which was substantiated in the context of a specific embodiment of the advocated system concept, namely with the aid of a burst-by-burst adaptive video transceiver.

Specifically, the main advantage of the described burst-by-burst adaptive transceiver technique is that irrespective of the prevailing channel conditions, the transceiver achieves always the best possible source-signal representation quality - such as video, speech or audio quality - by automatically adjusting the achievable bitrate and the associated multimedia source-signal representation quality in order to match the channel quality experienced. This is achieved on a near-instantaneous or burst-by-burst adaptive basis under given propagation conditions in order to cater for the effects of path-loss, fast-fading, slow-fading, dispersion, co-channel interference, etc. Furthermore, when the mobile is roaming in a hostile out-doors - or even hilly terrain - propagation environment, typically low-order, low-rate modem modes are invoked, while in benign indoor environments predominantly the high-rate, high source-signal representation quality modes are employed.

The described system embodiment has the following features:

1. A reliable instantaneous channel quality metric is employed, in order to appropriately configure the AQAM modem for maintaining the required target BER and the associated source signal representation quality. The range of potential channel quality metrics entails the pseudo-SNR, SINR, BER and its LLR-based channel estimates.

2. The perceived channel quality determines the number of bits that can be transmitted in a given transmitted packet or burst, which in turn predetermines the number of bits to be generated by the associated multimedia source codec, such as for example the associated video, audio, speech or handwriting codec. Hence the multimedia source codec has to be capable of adjusting the number of bits generated under the instruction of the burst-by-burst adaptive transceiver.

3. The transmitter mode requested by the receiver, in order to achieve the target performance has to be signalled by the receiver to the remote transmitter. Another scenario is, where the uplink and downlink channel quality is sufficiently similar for allowing the receiver to judge, what transmission mode the associated transmitter should use, in order for its transmitted signal to maintain the required transmission integrity.. Lastly, the mode of operation used by the transmitter can also be detected using blind detection techniques, for example in conjunction with the associated channel

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CLAIMS

1. A receiver unit comprising:

a burst-by-burst adaptive equalizer having an input for receiving data bursts from a communication channel, each burst containing a number of bits per symbol;

a computational unit for computing a received signal quality metric related to a bit error rate experienced during transmission over the communication channel;

an output for relaying the signal quality metric, conveying signal quality as perceived by the receiver unit, for use in determining a configuration for subsequent transmission bursts.

2. A receiver unit according to claim 1, wherein the received signal quality metric is evaluated from an interference parameter.

3. A receiver unit according to claim 2, wherein the signal quality metric is evaluated using channel impulse response estimates derived from a training sequence embedded in each transmission burst.

4. A receiver unit according to claim 2, wherein the interference parameter is a measure of co-channel interference.

5. A receiver unit according to claim 2, wherein the interference parameter is a measure of inter-symbol interference.

12. A receiver unit according to any one of the preceding claims, wherein the configuration defines the number of bits per symbol in each transmission burst, which is varied according to the signal quality metric computed from a previous transmission burst, as supplied by the output.

13. A receiver unit according to any one of the preceding claims, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, through the communication channel for configuration of the remote transmitter for subsequent transmission bursts, thereby to provide closed-loop feedback.

14. A receiver unit according to any one of claims 1 to 12, wherein the output is arranged to relay the signal quality metric, representative of signal quality as perceived by the receiver unit, to a transmitter unit local to the receiver unit for configuration of the local transmitter unit for subsequent transmission bursts to a remote receiver unit associated with the remote transmitter unit, thereby to provide open-loop feedback.

15. A receiver unit according to any one of claims 1 to 12, wherein the signal quality metric is internally used in a blind detection scheme to reconfigure the receiver unit for decoding subsequent transmission bursts.

16. A system comprising a receiver unit according to any one of claims 1 to 12 in combination with a transmitter unit, wherein the transmitter unit has an input connected to the output of the receiver unit for receiving the signal quality metric, the transmitter unit having a configuration that is responsive to the signal quality metric.

17. A system according to claim 16, wherein the transmitter unit comprises an interactive multimedia encoder having a configuration that is responsive to the signal quality metric.

18. A system according to claim 16 or 17, wherein the transmitter unit comprises a modem having a configuration that is responsive to the signal quality metric.

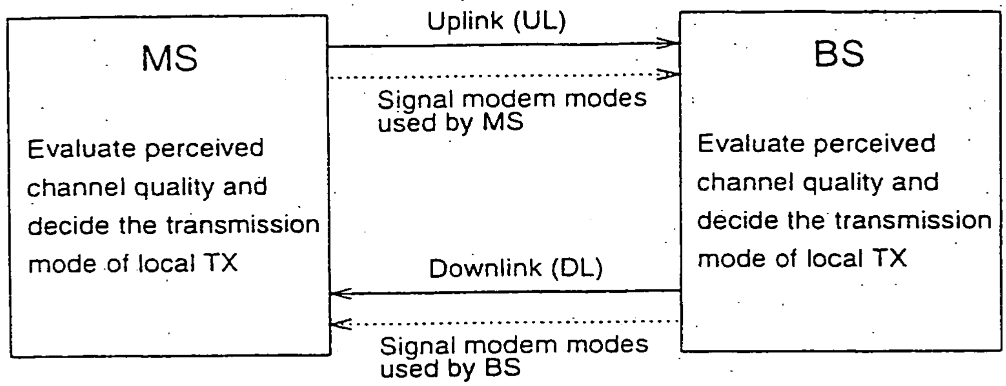

Figure 1(a)

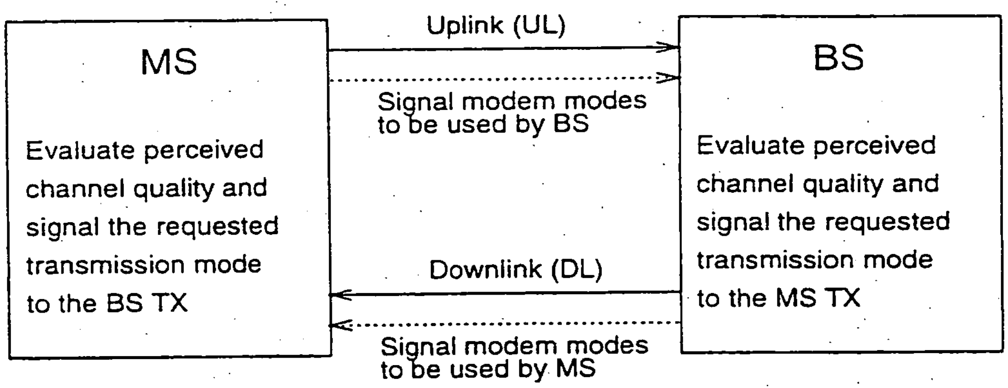

Figure 1(b)

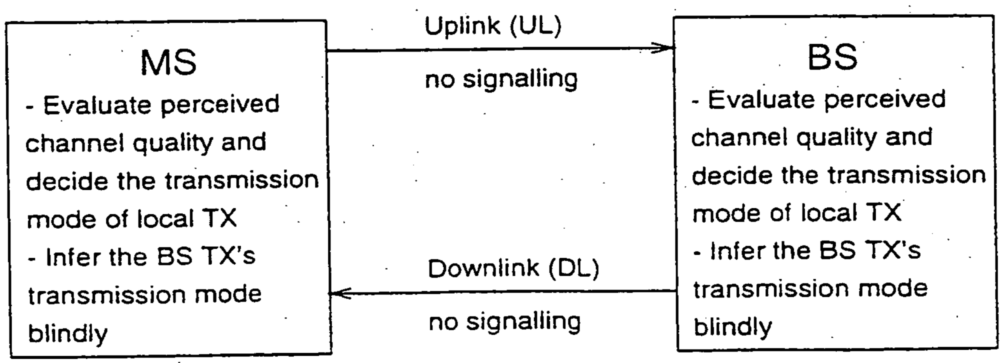

Figure 1(c)

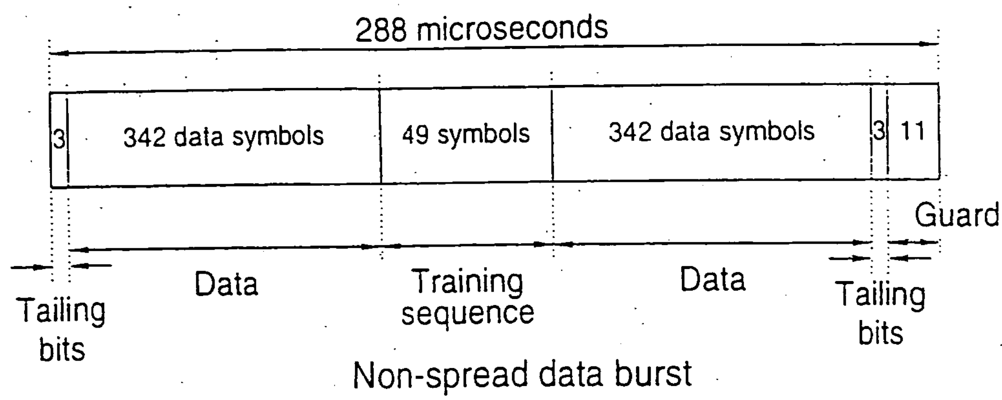

Figure 4

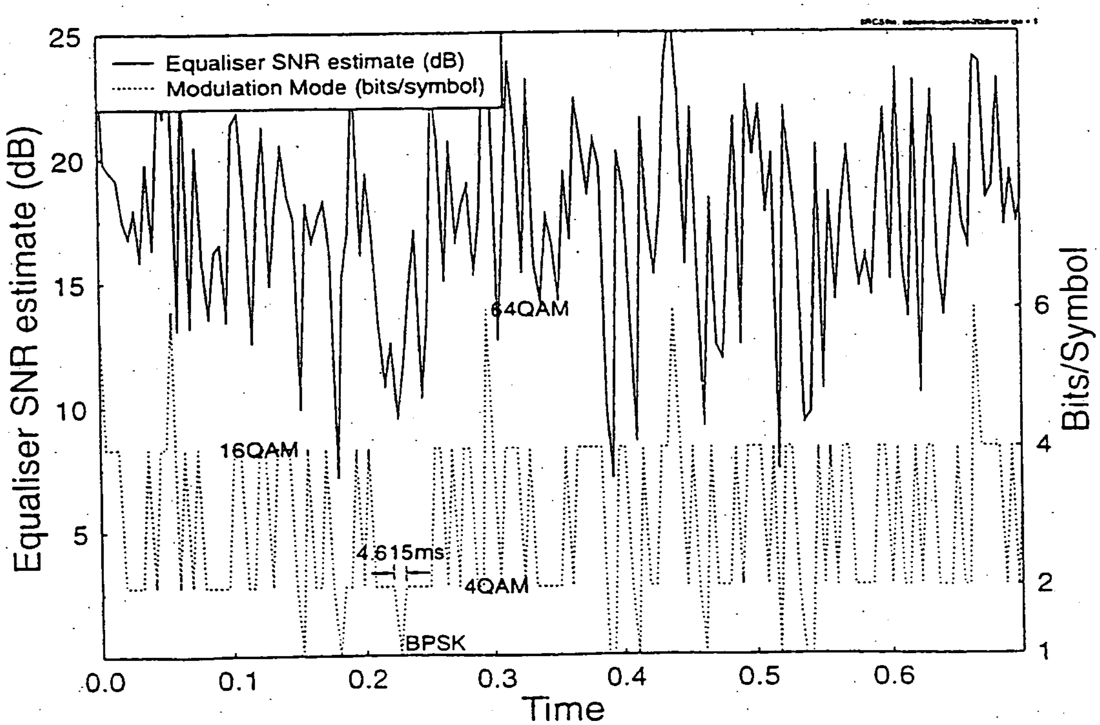

Figure 5

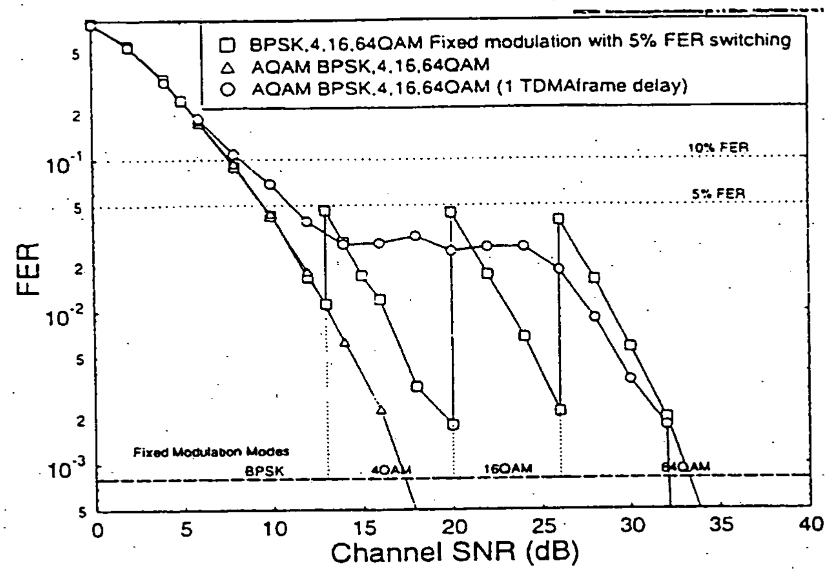

Figure 8

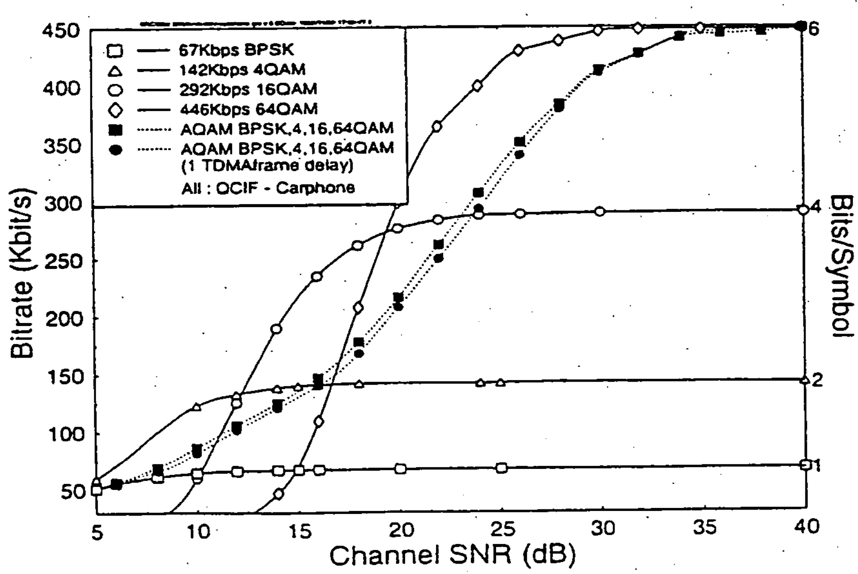

Figure 9

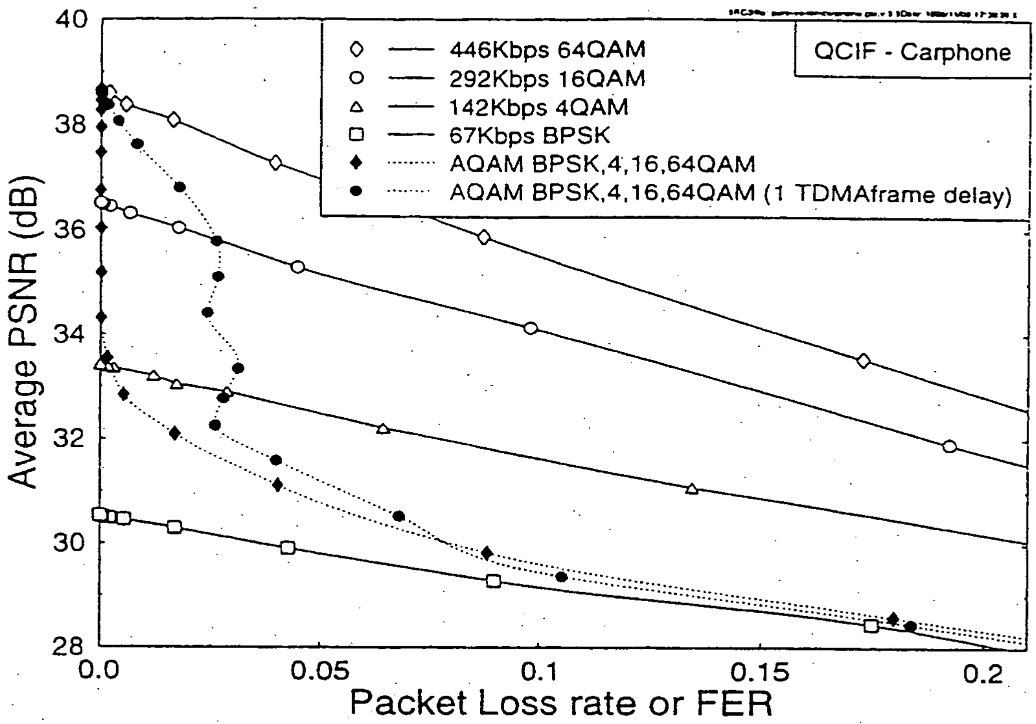

Figure 12

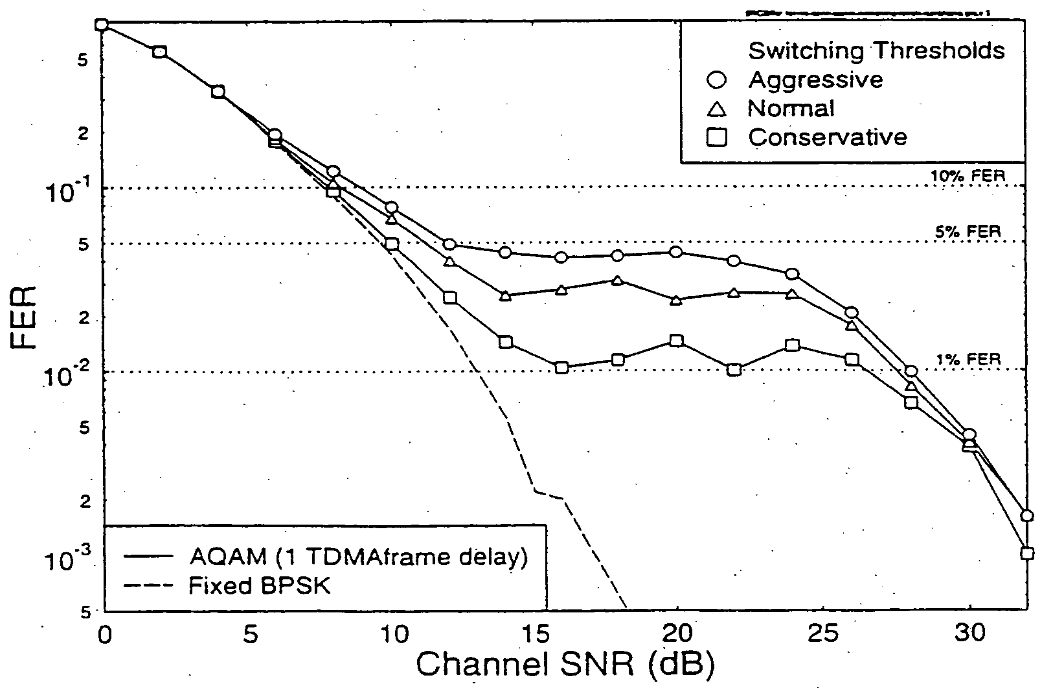

Figure 13

INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 00/00017

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 H04B1/10

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 H04B

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 98 51111 A (KONINKL PHILIPS ELECTRONICS NV ;PHILIPS AB (SE)) 12 November 1998 (1998-11-12) page 14, line 30 -page 16, line 31; figures 1,8 claims 6,14 page 5, line 30 -page 6, line 14 abstract —— -/—	1,7, 12-14, 16-18, 20,21

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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Patent document cited in search report		Publication date	Patent family member(s)		Publication date
WO 9851111	A	12-11-1998	EP	0935903 A	18-08-1999
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